ABSTRACT

In a method of coding speech signals transmitted to a user terminal during a VOIP telephone call set up via a packet transmission network the speech signals are conventionally divided into a succession of segments of the same duration by coders of the terminals before they are coded and transmitted in the form of packets and are reproduced from the packets received, eliminating any packet received twice and using a dissimulation algorithm for segments corresponding to missing packets. The method carries out an analysis during coding to identify any segment that is likely not to be able to be replaced by the dissimulation algorithm if the corresponding packet is missing. Any packet corresponding to a segment analyzed as likely not to be able to be replaced is transmitted twice by the sending terminal.